IN THE CLAIMS

1. (Original) A method, comprising:

providing a first signal from a first acoustic sensor and a second signal from a second acoustic sensor spaced apart from the first acoustic sensor, the first signal and the second signal each corresponding to two or more acoustic sources, said acoustic sources including a plurality of interfering sources and a desired source;

localizing the interfering sources from the first and second signals to provide a corresponding number of interfering source signals each corresponding to a different one of the interfering sources and each including a plurality of frequency components, the components each corresponding to a different frequency; and

suppressing one or more different frequency components of each of the interfering source signals to reduce noise.

2. (Original) The method of claim 1, wherein said suppressing includes extracting a desired signal representative of the desired source.

3. (Original) The method of claim 2, wherein said extracting includes determining a minimum value as a function of the interfering signals.

4. (Previously presented) The method of claim 1, wherein said localizing includes filtering with a number of coincidence patterns each corresponding to one of a number of

predetermined spatial positions relative to the first and second sensors, the patterns each providing phantom position information that varies with frequency relative to the one of the predetermined spatial positions.

5. (Original) The method of claim 1, further comprising delaying the first and second signals with a different dual delay line for each of a number of frequencies to provide a corresponding number of delayed signals to perform said localizing.

6. (Original) The method of claim 5, further comprising processing the delayed signals after said localizing to perform said suppressing.

7. (Original) The method of claim 6, further comprising:

transforming the first and second signals from a time domain form to a frequency domain form in terms of the frequencies before said delaying;

extracting a desired signal representative of the desired source, said extracting including said suppressing;

transforming the desired signal from a frequency domain form to a time domain form; and

generating an acoustic output representative of the desired source from the time domain form of the desired signal.

8. (Original) The method of claim 5, wherein the interfering signals are each determined from a unique pair of the delayed signals as a ratio between a difference in magnitude of the

unique pair of the delayed signals and a difference determined as a function of an amount of delay associated with each member of the unique pair of the delayed signals.

9. (Original) A system, comprising:

a pair of spaced apart acoustic sensors each arranged to detect two or more differently located acoustic sources and correspondingly generate a pair of input signals, said acoustic sources including a desired source and a plurality of interfering sources;

a delay operator responsive to said input signals to generate a number of delayed signals therefrom;

a localization operator responsive to said delayed signals to localize said interfering sources relative to location of said sensors and provide a plurality of interfering source signals each representative of a corresponding one of said interfering sources, said interfering source signals each being represented in terms of a plurality of frequency components, said components each corresponding to a different frequency;

an extraction operator responsive to said interfering source signals to suppress at least one of said frequency components of each of said interfering source signals and extract a desired signal corresponding to said desired source, said at least one of said frequency components being different for each of said interfering source signals; and

an output device responsive to said desired signal to provide an output corresponding to said desired source.

10. (Original) The system of claim 9, wherein said localization operator includes a filter to localize said interfering sources relative to a number of positions, said filter being based on a different coincidence pattern of ambiguous positional information that varies with frequency for each of said positions.

11. (Original) The system of claim 9, further comprising:

an analog-to-digital converter responsive to said input signals to convert each of said input signals from an analog form to a digital form;

a first transformation stage responsive to said digital form of said input signals to transform said input signals from a time domain form to a frequency domain form in terms of a plurality of discrete frequencies, said delay operator including a dual delay line for each of the frequencies;

a second transformation stage responsive to said desired signal to transform said desired signal from a digital frequency domain form to a digital time domain form; and

a digital-to-analog converter responsive to said digital time domain form to convert said desired signal to an analog output form for said output device.

12. (Previously presented) The system of claim 9, wherein said delay operator, said localization operator, and said extraction operator are provided by a solid state signal processing device.

13. (Previously presented) The system of claim 9, wherein said desired source signal is determined as a function of said interfering signals.

14. (Previously presented) The system of claim 9, wherein said interfering source signals are each determined from a unique pair of said delayed signals.

15. (Original) The system of claim 14, wherein said interfering signals each correspond to a ratio between a difference in magnitude of said unique pair of said delayed signals and a difference determined as a function of an amount of delay associated with each member of said unique pair of said delayed signals.

16. (Previously presented) The system of claim 9, wherein said output device is configured to provide an acoustic output representative of said desired source.

17. (Original) A method, comprising:

positioning a first acoustic sensor and a second acoustic sensor to detect a plurality of differently located acoustic sources;

generating a first signal corresponding to said sources with said first sensor and a second signal corresponding to said sources with said second sensor;

providing a number of delayed signal pairs from the first and second signals, the delayed signal pairs each corresponding to one of a number of positions relative to the first and second sensors; and

localizing the sources as a function of the delayed signal pairs and a number of coincidence patterns, the patterns each corresponding to one of the positions and establishing an expected variation of acoustic source position information with frequency attributable to a source at the one of the positions.

18. (Original) The method of claim 17, wherein the coincidence patterns each correspond to a number of relationships characterizing a variation of phantom acoustic source position with frequency, the relationships each corresponding to a different ambiguous phase multiple.

19. (Original) The method of claim 18, further comprising determining the relationships for each of the coincidence patterns as a function of distance separating the first and second sensors.

20. (Original) The method of claim 18, wherein the relationships each correspond to a secondary contour that curves in relation to a primary contour, the primary contour representing frequency invariant acoustic source position information determined from the delayed signal pair corresponding to the one of the positions.

21. (Previously presented) The method of claim 17, wherein said localizing includes filtering with the coincidence patterns to enhance true position information with phantom position information.

22. (Original) The method of claim 21, wherein said localizing includes integrating over time and integrating over frequency.

23. (Previously presented) The method of claim 17, wherein the first sensor and second sensor are part of a hearing aid device and further comprising adjusting the delayed signal pairs with a head-related-transfer function.

24. (Previously presented) The method of claim 17, further comprising:

extracting a desired signal after said localizing; and

suppressing a different set of frequency components for each of a selected number of the sources to reduce noise.

25. (Previously presented) The method of claim 17, wherein the positions each correspond to an azimuth established relative to the first and second sensors and further comprising generating a map showing relative location of each of the sources.

26. (Original) A system, comprising:

a pair of spaced apart acoustic sensors each configured to generate a corresponding one of a pair of inputs signals, the signals being representative of a number of differently located acoustic sources;

a delay operator responsive to said input signals to generate a number of delayed signals each corresponding to one of a number of positions relative to said sensors;

a localization operator responsive to said delayed signals to determine a number of sound source localization signals from said delayed signals and a number of coincidence patterns, said patterns each corresponding to one of said positions and relating frequency varying sound source position information caused by ambiguous phase multiples to said one of said positions to improve sound source localization; and

an output device responsive to said localization signals to provide an output corresponding to at least one of said sources.

27. (Original) The system of claim 26, further comprising:

an analog-to-digital converter responsive to said input signals to convert each of said input signals from an analog form to a digital form; and

a first transformation stage responsive to said digital form of said input signals to transform said input signals from a time domain form to a frequency domain form in terms of a plurality of discrete frequencies, said delay operator including a dual delay line for each of the frequencies.

28. (Original) The system of claim 27, further comprising:

an extraction operator responsive to said localization signals to extract a desired signal;

a second transformation stage responsive to said desired signal to transform said desired signal from a digital frequency domain form to a digital time domain form; and

a digital to analog converter responsive to said digital time domain form to convert said desired signal to an analog output form for said output device.

29. (Previously presented) The system of claim 26, wherein said output device is configured to provide a map of acoustic source locations.

30. (Previously presented) The system of claim 26, wherein said delay operator and said localization operator are defined by an integrated solid state signal processor.

31. (Previously presented) The system of claim 26, wherein said localization operator responds to said delay signals to determine a closest one of said positions for one of said sources as a function of at least one of said delayed signals corresponding to said closest one of said positions and at least two other of said delayed signals corresponding to other of said positions, said at least two other of said delayed signals being determined with a corresponding one of said coincidence patterns.

32. (Original) A system, comprising:

a pair of spaced apart acoustic sensors each generating a corresponding one of a pair of inputs signals, the signals each being representative of a number of differently located sound sources;

a signal processor responsive to said sensors, said processor including: (a) a means for providing a number of delayed signals from said input signals, the delayed signals each corresponding to one of a number of positions relative to said first and second sensors; (b) a means for localizing each of said sound sources to one of said positions as a function of said delayed signals and a corresponding one of a number of patterns of frequency invariant data

corresponding to one of said positions and frequency dependent data corresponding to at least two other of said positions; (c) a means for suppressing a different frequency component of each of a selected number of said sources causing interference and for extracting a desired signal representative of one of said sources; and

an output device responsive to said desired signal to provide an output corresponding to said one of said sources.

33. (Original) The system of claim 32, wherein said processor includes a means for adjusting said delayed signals with a head-related-transfer-function.

34. (Original) A signal processing system, comprising:

(a) a first sensor at a first location configured to provide a first signal corresponding to an acoustic signal, said acoustic signal including a desired signal emanating from a selected source and noise emanating from a noise source;

(b) a second sensor at a second location configured to provide a second signal corresponding to said acoustic signal;

(c) a signal processor responsive to said first and second signals to generate a discrete first spectral signal corresponding to said first signal and a discrete second spectral signal corresponding to said second signal, said processor being configured to delay said first and second spectral signals by a number of time intervals to generate a number of delayed first signals and a number of delayed second signals and provide a time increment signal, said time increment signal corresponding to separation of the selected source from the noise source, and

said processor being further configured to generate an output signal as a function of said time increment signal; and

(d) an output device responsive to said output signal to provide an output representative of said desired signal.

35. (Original) The system of claim 34, wherein said first and second sensors each include a microphone and said output device includes an audio speaker.

36. (Original) The system of claim 34, wherein said processor includes an analog to digital conversion circuit configured to provide said discrete first spectral signal.

37. (Original) The system of claim 34, wherein generation of said first and second spectral signals includes execution of a discrete Fourier transform algorithm.

38. (Original) The system of claim 34, wherein said first and second sensors are configured for movement to select said desired signal in accordance with position of said first and second sensors, said first and second sensors being configured to be spatially fixed relative to each other.

39. (Previously presented) The system of claim 34, wherein each of said delayed first signals corresponds to one of a number of first taps from a first delay line, and each of said delayed second signals corresponds to one of a number of second taps from a second delay line.

40. (Original) The system of claim 39, wherein determination of said output signal corresponds to:

said first and second delay lines being configured in a dual delay line configuration;

said discrete first spectral signal being input to said first delay line and said discrete second spectral signal being input to said second delay line; and

each of said first taps, said second taps, and said first and second spectral signals being arranged as a number of signal pairs, said signal pairs including a first portion of signal pairs and a second portion of signal pairs, said processor being configured to perform a first operation on each of said signal pairs of said first portion as a function of said time intervals, said processor being configured to perform a second operation on each of said signal pairs of said second portion as a function of said time intervals, said first operation being different from said second operation.

41. (Currently amended) A method of signal processing, comprising:

(a) positioning a first and second sensor relative to a first signal source, the first and second sensor being spaced apart from each other, and a second signal source being spaced apart from the first signal source;

(b) providing a first signal from the first sensor and a second signal from the second signal, the first and second signals each being representative of a composite acoustic signal including a desired signal from the first signal source and an unwanted signal from the second signal source;

(c) establishing a number of spectral signals from the first and second signals as a function of a number of frequencies, each of the spectral signals representing a different position relative to the first signal source;

(d) determining a member of the spectral signals representative of position of the second signal source; and

(e) generating an output signal from the member, the output signal being representative of spectral content of the desired signal from the first signal source.

42. (Original) The method of claim 41, wherein the member is determined as a function of a phase difference value.

43. (Original) The method of claim 41, wherein the desired signal includes speech and the output signal is provided by a hearing aid device.

44. (Previously presented) The method of claim 41, further comprising repositioning the first and second sensors to extract a third signal from a third signal source.

45. (Previously presented) The method of claim 41, wherein said establishing includes:

(a1) delaying each of the first and second signals by a number of time intervals to generate a number of delayed first signals and a number of delayed second signals; and

(a2) comparing each of the delayed first signals to a corresponding one of the delayed second signals, each of the spectral signals being a function of at least one of the delayed first signals and the delayed second signals.